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6 OBSERVATIONS OF A QUASI-PERIODIC INSTABILITY IN A LINEAR PREDICTION ANALYSIS OF VOICED SPEECH •

10 D.G. NICHOL[redacted] R.E. BOGNER

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OBSERVATIONS OF A QUASI-PERIODIC INSTABILITY IN A LINEAR PREDICTION ANALYSIS OF VOICED SPEECH

D.G. Nichol and R.E. Bogner*

SUMMARY

A significant semiperiodic fluctuation of the vocal tract area functions derived by linear prediction of the speech waveform has been noted during apparently stationary voiced segments of speech. In one example some values of the area function varied over a range of 9:1 over a few pitch-periods. The phenomenon is attributed to 'beating' of the pitch-period and the time interval between successive computations which causes variations of the time relationship between glottal pulse and analysis window. This is supported by the fact that no fluctuations occur in the area function derived from natural or synthetic speech when the computation interval is equal to the pitch period. Any slight difference between the two leads to significant pulsations however. A simple theoretical model is used to show how the positioning of the analysis window can influence area function estimates.

The problem can be largely overcome by using longer time windows (greater than 2.5 pitch periods), or alternatively by averaging the area functions over several adjacent intervals.

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A significant semiperiodic fluctuation of the vocal tract area functions derived by linear prediction of the speech waveform has been noted during apparently stationary voiced segments of speech. In one example some values of the area function varied over a range of 9:1 over a few pitch-periods. The phenomenon is attributed to 'beating' of the pitch-period and the time interval between successive computations which causes variations of the time relationship between glottal pulse and analysis window. This is supported by the fact that no fluctuations occur in the area function derived from natural or synthetic speech when the computation interval is equal to the pitch period. Any slight difference between the two leads to significant pulsations however. A simple theoretical model is used to show how the positioning of the analysis window can influence area function estimates.

The problem can be largely overcome by using longer time windows (greater than 2.5 pitch periods), or alternatively by averaging the area functions over several adjacent intervals.

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1. INTRODUCTION

Several attempts have been made recently to use linear prediction analysis of speech for isolated word(ref.1,2) and spoken digit recognition(ref.3,4). The feature chosen for the recognition algorithm in these studies was the set of linear prediction coefficients. It is well known that an estimate of the vocal tract area function can be derived from these coefficients(ref.5,6,7) and the present paper arose from a study of the usefulness of this function for both speech and voice recognition. Because of the extensive information available from phonetic and articulatory studies of speech production it was believed that the vocal tract area function (VTAF) would be an advantageous feature for the pattern recognition process. To test this idea, and also to compare the various formulations of the linear prediction models, it was decided to display the VTAF as an intensity-modulated picture of vocal tract position versus time, with the area plotted as a grey-level. This is of course a similar display to the well known spectrogram. We shall refer to these displays as VTAF pictures.

Figure 1 shows a typical VTAF picture produced from real speech. The most obvious feature of this picture is the strong pulsations seen, for example, at the intervals labelled 2, 5 and 8. It is believed these pulsations are artifacts of the analysis as no evidence of them is apparent in the time series or spectrogram.

Before discussing this phenomenon in more detail we describe briefly the production of the pictures.

2. EXPERIMENTAL RESULTS

2.1 The VTAF picture

The first linear prediction model used in this study was that due to Wakita(ref.7). This is so-called auto-correlation technique and was chosen because, for non pitch-synchronous analysis, these formulations are generally more stable and robust than the "covariance" methods although for pitch-synchronous analysis the latter are capable of giving better estimates of the actual vocal tract(ref.8,9).

Suppose that the anti-aliasing filtered speech signal is sampled at frequency $f_s = 1/T$, and that n_W samples are included in each autocorrelation window and that a new computation of the VTAF is made every n_c samples. If m_1 linear prediction coefficients are used then $m_V = m_1 + 1$ vocal tract areas are produced at time intervals of t_c where

$$t_c = n_c T \tag{1}$$

Denoting the array of vocal tract areas $a_i(t)$ obtained at time t as a vector $\underline{\alpha}(t)$ we have

$$\alpha(t) = a_1(t), a_2(t), a_3(t) \ldots . a_{m_V}(t) \tag{2}$$

In n successive estimates of $\underline{\alpha}(t)$ are evaluated then the resulting sets of these $\underline{\underline{\alpha}}(t)$ may be regarded as an $(m \times n)$ matrix

$$\underset{\sim}{A} = \begin{bmatrix} a_{11} & a_{12} & a_{1n} \\ a_{21} & a_{22} & a_{2n} \\ . & & . \\ . & & . \\ . & & . \\ a_{m1} & & a_{mn} \end{bmatrix} \tag{3}$$

where we have written a_{mn} for $a_m(nT)$.

This matrix can be plotted as an (m x n) digital picture where the grey levels are assigned by some mapping from the values of the elements of $\underset{\sim}{A}$ to the set of grey levels.

To produce figure 1 the values $f_s = 8192$ Hz, $m_V = 9$, $n_w = 64$, $n_c = 64$ and $n = 1024$ were used. Now a 9 x 1024 picture is a very cumbersome shape and so this was split into eight 9 x 128 subpictures, which for display purposes were interpolated (by a two-dimensional Fast Fourier Transform) into eight 36 x 512 subpictures. These eight subpictures were plotted, one below the other, as in figure 1 on an intensity-modulated CRT. The bottom of each subpicture represents the glottis and the top the lips. The grey levels have been assigned such that the larger the area the greater the whiteness. Thus the point of maximum constriction is the darkest region in each column. It should be noted that the Wakita model assumes a constant glottis area and thus the lower edge of each picture is a constant grey level. Some regions of the picture are blank. This is due to use of an energy-detecting algorithm which assigns arbitrary zero levels to the VTAF's when the total signal occurring in the time series window is below a given threshold (as during silences between utterances). Each subpicture represents 1 s of real time and thus 8 s is shown overall.

2.2 Observations

In figure 1 the occurrence of periods of pulsations is easily observed; the most obvious of these are indicated in the picture. The utterance shown in this picture is the phrase "Speak to me now, bad kangaroo!" repeated three times by an Australian female speaker. The observed pulsations occur during constant vowel segments where little or no real change in the vocal tract is occurring.

This is supported by an examination of the speech time series corresponding to the utterance. Figure 2(a) shows the speech waveform corresponding to the /ae / in 'bad' which for this speaker is remarkably stationary. The corresponding VTAF's are plotted in figure 2(b) and are clearly fluctuating, an effect which does not auger well for using the VTAF's in any automatic speech recognition process. The plots in figure 2(b) are in fact the square-root of the VTAF and thus the estimates of area actually vary by the order of 9:1 during this apparently stationary segment.

We had not observed this phenomenon previously, even though several VTAF pictures of Australian male speakers repeating the same phrase had been made. This suggested that the phenomenon may be sensitive to pitch period.

Now for linear prediction models of the Wakita type the analysis begins by windowing the time series by a Hanning weighting function. The only parameter which is changing during a stationary segment is thus the position of the glottal pulse within the Hanning window (unless the analysis is pitch-synchronous). We shall now examine this effect and show it can cause the observed phenomenon.

3. ANALYSIS OF WINDOW POSITION EFFECTS

We examine the effect of the time relationship between the autocorrelation window and the speech waveform (figure 3). To facilitate analysis we use a model comprising a second order (two junction, three section) vocal tract yielding an impulse response of the form

$$h(n) = r^n \cos n\omega T, n = 0,1,2 \ldots . \quad (4)$$

We use a window of the form

$$w(n) = ½ + ½ \cos \frac{2\pi(n-d)}{N}, n = -\frac{N}{2} \text{ to } +\frac{N}{2} \quad (5)$$

where d is the delay whose effect is of interest, and N the duration of the window is large compared with $1/(r-1)$, i.e. the interval over which the impulse response has significant magnitude. The excitation is taken to be a unit pulse, and thus the model speech signal s(n) is the same as h(n).

This model is not realistic, but it does aid our appreciation of effects which can arise, and yields a sufficient explanation of our observations.

Figure 3 shows effects on the windowed signal $s_w(n)$ variation of the delay. Two cases are shown, viz.

(a) Centred window, in which $w(n) \doteq 1$ over the effective duration of $s(n)$, i.e. we have $s_w(n) \doteq s(n)$

(b) Delayed window, in which the curved rise of $w(n)$ progressively magnifies the signal, producing a compensation of the damping of $s(n)$.

Figure 4 shows for example the shape of the window

$$w_d(n) = 0.5[1+\cos\frac{2\pi}{128}(n-61)] \qquad (6)$$

compared with the expotential Rr_1^n with $R = 0.014$ and $r_1 = 1.26$, which were chosen in an ad hoc manner simply for demonstration purposes. The expotential appears to be a reasonable approximation to $w_d(n)$.

We see that for $s(n)$ of the form

$$s(n) = r_s^n \cos n\omega T \qquad (7)$$

with $r < 1$ the delayed window would cause $s_w(n)$ to be approximately

$$s_w(n) = s(n)\, w_d(n) \doteq R\,(r_s r_1)^n \cos n\omega T \qquad (8)$$

Now, for speech sampled at 10 000 s^{-1}, the value of r_s is likely to be in the range 0.985 to 0.9 (i.e. approximately 50 Hz to 300 Hz formant bandwidth respectively). Thus, the apparent value of the ratio corresponding to r_s in $s(n)$ as given by (7) becomes the value $r_s r_1$ in (8) and the latter may be grossly in error, and even exceed unity as with the example values of $r_1 = 1.26$ and $r_s = 0.9$.

Next we study the effect of a discrepancy in the value of r_s on the area function of a model vocal tract.

Figure 5 shows an acoustic tube (or transmission line model of a vocal tract in which there are three sections of area a_m, $m = 0$ at the lips end, 1 for the middle section and 2 at the glottis. There are thus two junctions whose volume velocity reflection coefficients μ_m, $m = 1$ and 2 are given by (ref.10)

$$\mu_m = \frac{a_{m-1} - a_m}{a_{m-1} + a_m} \qquad (9)$$

The termination at the lips is assumed to be equivalent to a tube section of infinite area, resulting in a volume velocity reflection coefficient of -1. For convenience in analysis we associate all the delay (i.e. sum of delays for forward and backward travelling waves) with the backward travelling wave in each section. The physical model of figure 5(a) may then be represented by the signal flow model of figure 5(b).

Analysis of this model shows that the transfer function $H(z) = U_L(z)/U_G(z)$ is given by

$$H(z) = \frac{(1+\mu_1)(1+\mu_2)}{1+z^{-1}\,\mu_1\,(1+\mu_2) - z^{-2}\,\mu_2} \qquad (10)$$

$$= \frac{4\,a_0 a_1}{(a_0 + a_1)(a_1 + a_2)} \cdot \frac{1}{1+z^{-1}\left(\frac{2a_1}{a_1 + a_2}\right)\left(\frac{a_0 - a_1}{a_0 + a_1}\right) + z^{-2}\left(\frac{a_2 - a_1}{a_2 + a_1}\right)} \qquad (11)$$

The impulse response of this system is of the form

$$h(n) = h(0)\, r^n \cos n\omega T \tag{12}$$

where

$$h(0) = (1+\mu_1)(1+\mu_2) = \frac{4a_0\, a_1}{(a_0+a_1)(a_1+a_2)}\,, \tag{13}$$

$$r^2 = -\mu_2 = \frac{a_2 - a_1}{a_2 + a_1}\,, \tag{14}$$

and

$$\cos \omega T = \frac{1}{2r}\mu_1(1+\mu_2) = \frac{1}{2r}\left(\frac{2a_2}{a_1+a_2}\right)\left(\frac{a_0-a_1}{a_0+a_1}\right). \tag{15}$$

From (14) we see that, for this two junction model, the damping ratio r depends only on the reflection coefficient of the junction closest to the glottis, i.e. on the area ratio at this junction. We might query the physical meaning of the possibility that $a_2 - a_1 < 0$ i.e., $r^2 < 0$ in (14). Detailed analysis shows that the impulse response is then not oscillatory, corresponds to real poles, and is not of interest in the present study.

To apply this two junction model to realistic speech parameters we set the length of each section equal to half the length of the vocal tract, i.e. about 9 cm. The tie in with the previous discussion of the 10 000 Hz sampling rate, it is convenient to let each of the sections be equivalent to an each way delay of 3×10^{-4} s. The delay T in the second order model described by equation (8) is thus 6×10^{-4} s, and the relevant values of r for use in these equations are $(r_s r_1)^6$ or $(r_s)^6$. Of course this change in fact replaces the second order system by a 12th order system if the original sampling rate is maintained, since denominator factors of H(z) in (12) of the form

$$(1 - z^{-1}\, r\, e^{j\omega T})$$

are replaced by factors of the form

$$(1 - z^{-6}\, r^6\, e^{j6\omega T}).$$

Each of these factors results in 6 poles, but the base pole of each is the same as previously, i.e. at $z = re^{j\omega T}$. From (14) we find

$$a_2 = \frac{1+r^2}{1-r^2}\, a_1 \tag{16}$$

and

$$\left.\frac{\frac{da_2}{a_2}}{dr}\right|_{a_1} = \frac{4r}{1-r^4}\,. \tag{17}$$

and

$$\left.\frac{\frac{da_1}{a_1}}{dr}\right|_{a_2} = \frac{-4r}{1-r^4}\,. \tag{18}$$

From (17) and also (18) we see that the proportional variation of either area a_1 or a_2 with r, while the other is fixed becomes very great as $r \to 1$. We found earlier that the effect of the delayed window on the apparent damping was sufficient to make r pass through unity, and thus the system may incur such great sensitivities. For example, varying r from 0.90 to 0.99 causes $\frac{a_2}{a_1}$ to change from 9.53 to 99.5. Clearly this effect is sufficient to account for variations as large as those observed in Section 2.2.

For completeness, we study the effect at the first junction. We note that ωT is not affected by the window delay phenomenon, and thus we set $d(\cos \omega T) = 0$ when differentiating (15).

We find

$$a_0 = a_1 \frac{r}{1 + r^2} \cos \omega T \tag{19}$$

and

$$\left. \frac{\frac{da_0}{a_0}}{dr} \right|_{a_1} = \frac{1 - r^2}{r(1 + r^2)} \tag{20}$$

which shows that the area ratios at the first junction are not strongly influenced by r.

Note also that there is a gross effect on the initial value h(0) or the windowed response. Via (13) we see that this can affect the product of the junction transmission coefficients i.e. $(1 + \mu_1)(1 + \mu_2)$. The effect on a particular feature however is not explicit.

For more complex vocal tract models, the effects are more complex, but we have demonstrated a sufficient mechanism to account for the observations. One previous study(ref.11) showed that under moderate variation of the pole dampings in a 5 pole signal, the resultant VTAF retained its gross features, but underwent a gradual smooth change.

4. EXPERIMENTAL DIAGNOSIS

To test these ideas, synthetic vowels were generated (in the computer) using an all-pole filter and known excitation function. Details of the synthesis algorithm used are given in Rogers(ref.12). The four poles used were derived from the values of formant positions and bandwidths given by Fant(ref.13) (Table 1).

TABLE 1. POLE POSITIONS IN TERMS OF FORMANT FREQUENCIES AND BANDWIDTHS (AFTER FANT(REF.13))

Vowel	First formant		Second formant		Third formant		Fourth formant	
	freq.	b-width	freq.	b-width	freq.	b-width	freq.	b-width
/ae/	616	57	1072	72	2430	130	3410	175
/e/	432	39	1959	95	2722	170	3500	325
/I/	222	60	2244	75	3140	240	3700	230
/ρ/	510	54	900	65	2400	100	3220	135
/N/	231	69	615	50	2375	110	3320	115

Figure 6 shows plots of the VTAF's obtained for the synthetic vowel /ae/ when the pitch period n_p has been made equal to the computation interval n_c. In each of the four columns however the 'phase' of the excitation function relative to the computation window is different (as indicated in the figure). Clearly the areas calculated vary with this phase. This means that when $n_p \neq n_c$ the area calculated from a constant waveform will fluctuate as the position of the excitation impulse changes within the window. Figure 7 shows this happening when $n_p = 0.8\, n_c$ for five different synthetic vowels.

The reason this effect had not been observed in previous VTAF pictures of male speakers is believed to be that the computation interval used (64 samples, equivalent to 7.81 mS at 8192 Hz sampling rate) is quite close to the pitch period of the speakers analysed. Thus, fluctuations are not observed as the excitation function remains in a nearly constant position in the Hanning window. It should be remembered however that the errors may still be present in the analysis but not show up as fluctuations. For the female speaker $n_p \approx 0.9\, n_c$ and the fluctuations are obvious (figure 1). This interpretation is supported by the fact that fluctuations did appear in male VTAF's that have been reprocessed with larger values of n_c.

5. SUPPRESSING THE FLUCTUATIONS

It appears from the above discussion that a partial cure for the problem of fluctuations would be to increase the size of the Hanning window used to estimate the autocorrelation function. This should improve the estimate of r_s. Figure 8 shows the synthetic vowel /ae) as shown in figure 6 but with $n_w = 3.5\, n_p$. We see that the variations are suppressed. To test this on real speech the VTAF picture (figure 1) was reprocessed with $n_w = 192$ (approximately $3.5\, n_p$) and the result is shown in figure 9. The fluctuations have indeed been largely suppressed.

6. CONCLUDING REMARKS

The autocorrelation methods of linear prediction have a certain attraction in terms of robustness and economy of computing effort. We have shown that care must be taken in choosing lengths for the analysis, but provided this is done then consistent estimates of the vocal tract area are obtained. If Hanning windows of length $> 2.5\, n_p$ are used, the resultant area functions appear to have the robustness desirable for automatic speech recognition, or for use in visual displays for speech training and phonetic studies.

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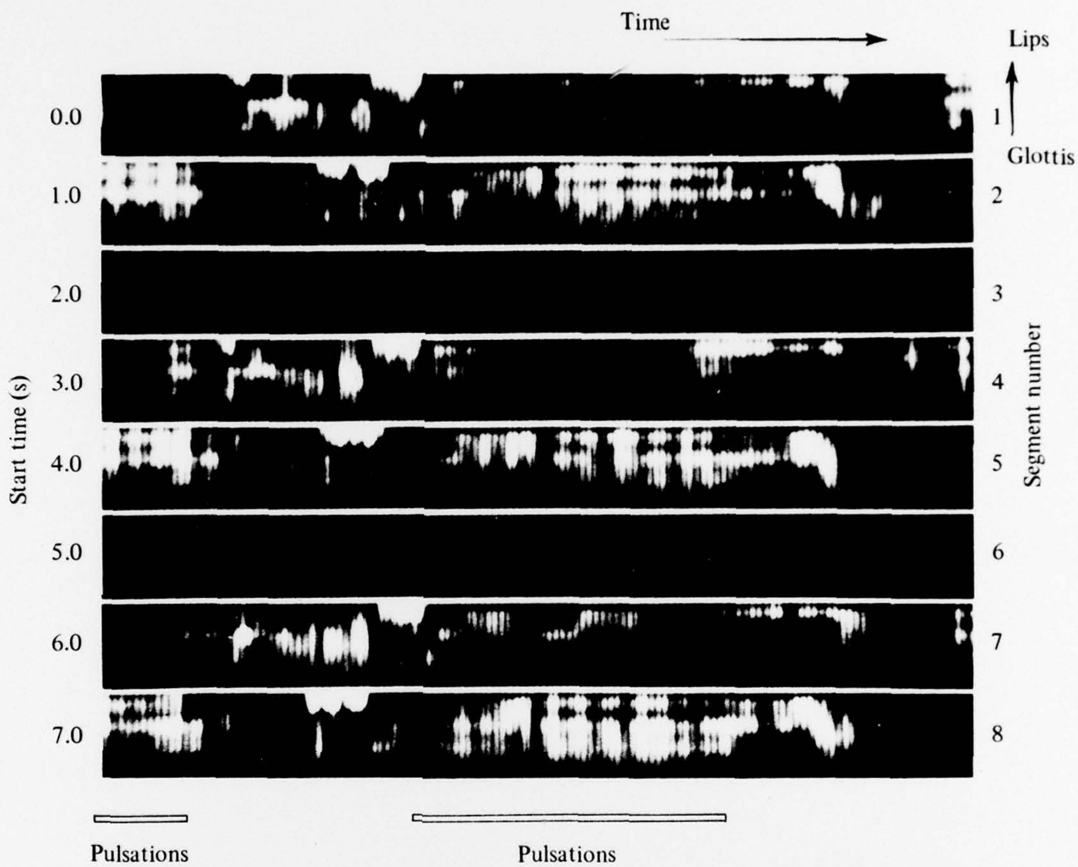

Figure 1. Vocal tract area function for sentence "Speak to me now, bad kangaroo!"

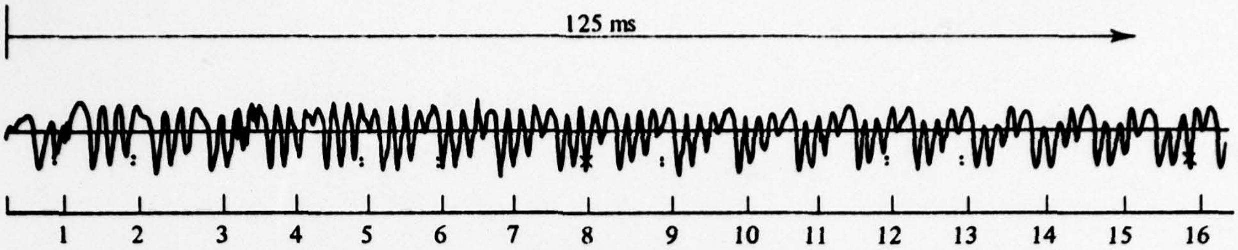

Figure 2(a). Speech waveform for /ae/

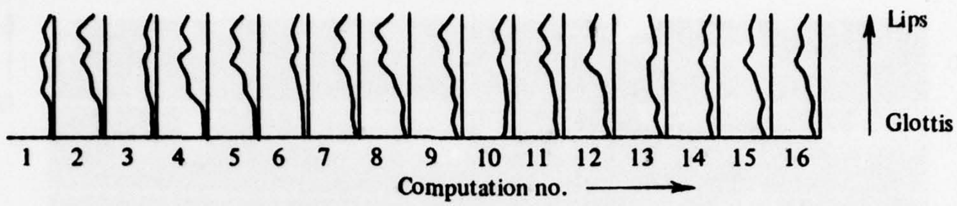

Figure 2(b). Vocal tract area function for /ae/

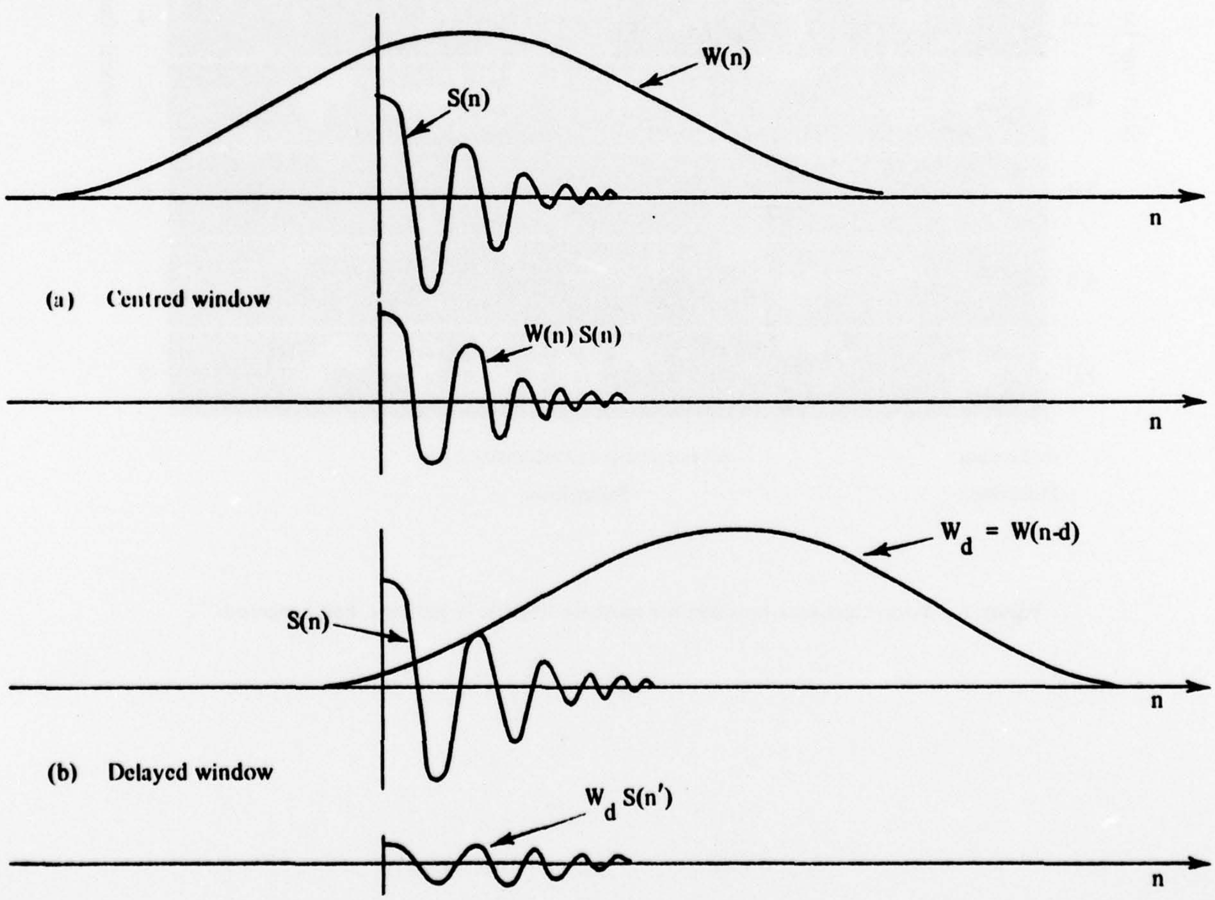

Figure 3. Effect on windowed signal $s_w(n)$ of relative shift between signal and window

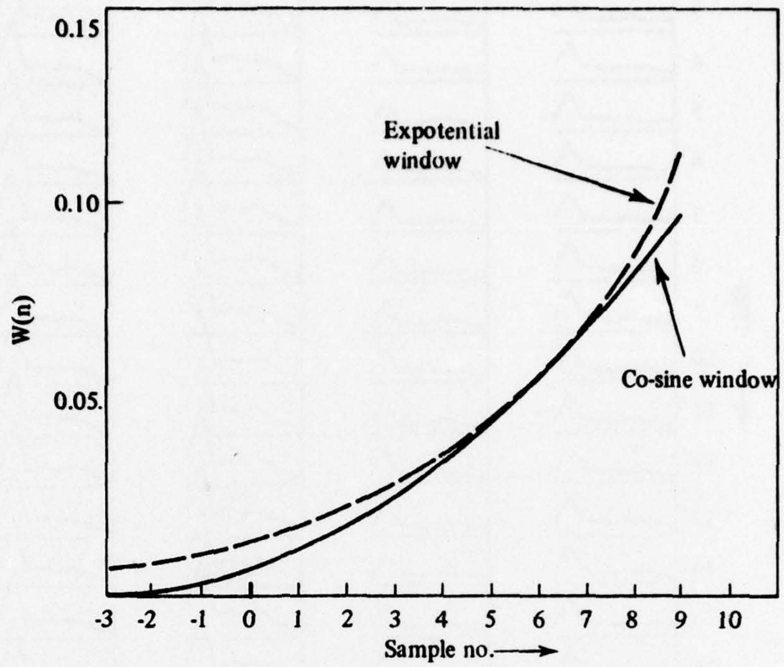

Figure 4. Approximation to a delayed window $w_d(n) = 0.58(1 + \cos\frac{2\pi}{128}(n\text{-}61))$ by a rising expotential 0.014×1.26^n

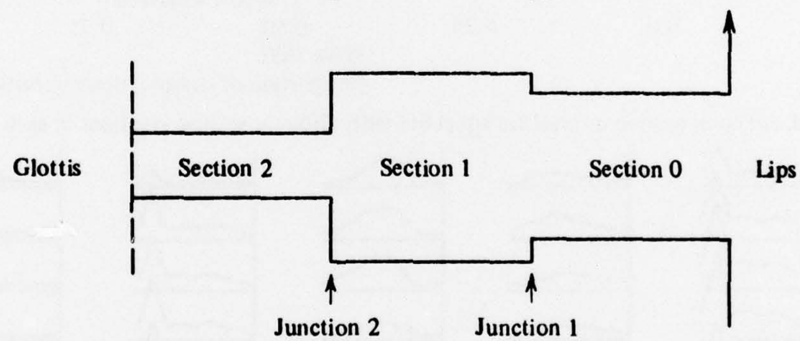

Figure 5(a). Physical tube model

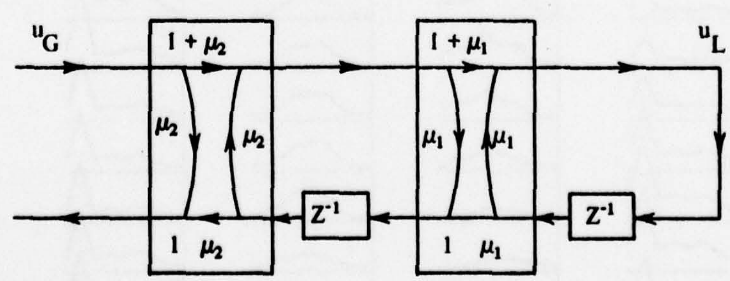

Figure 5(b). Signal flow model

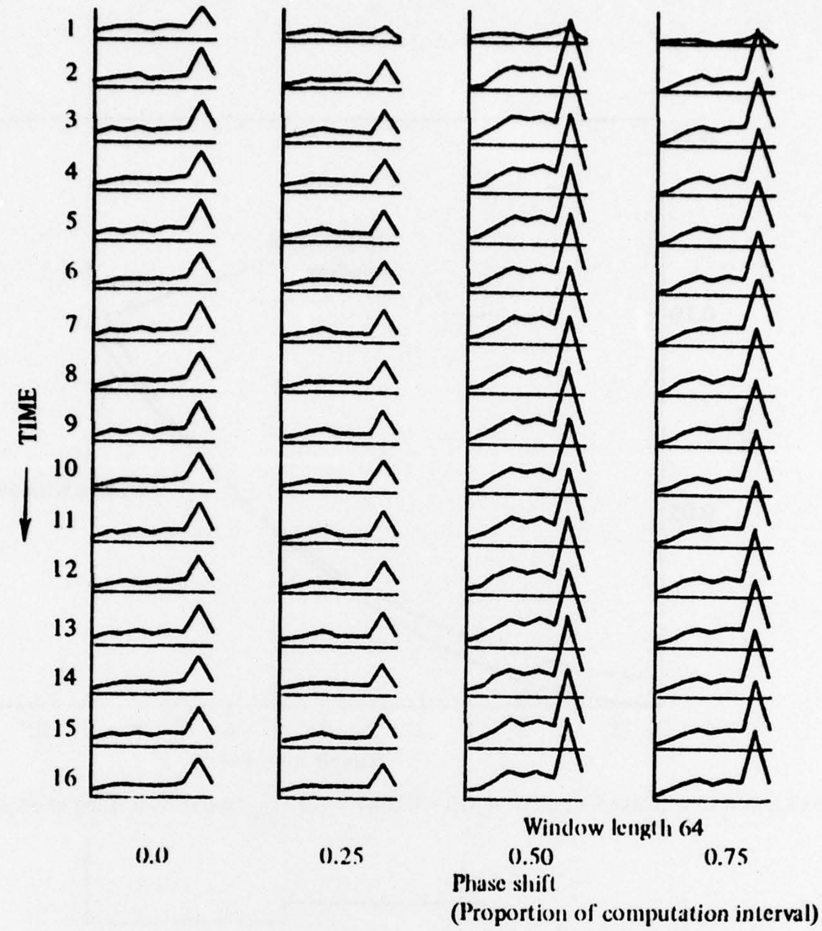

Figure 6. Pitch period and computation interval are equal but with different relative positions in each column

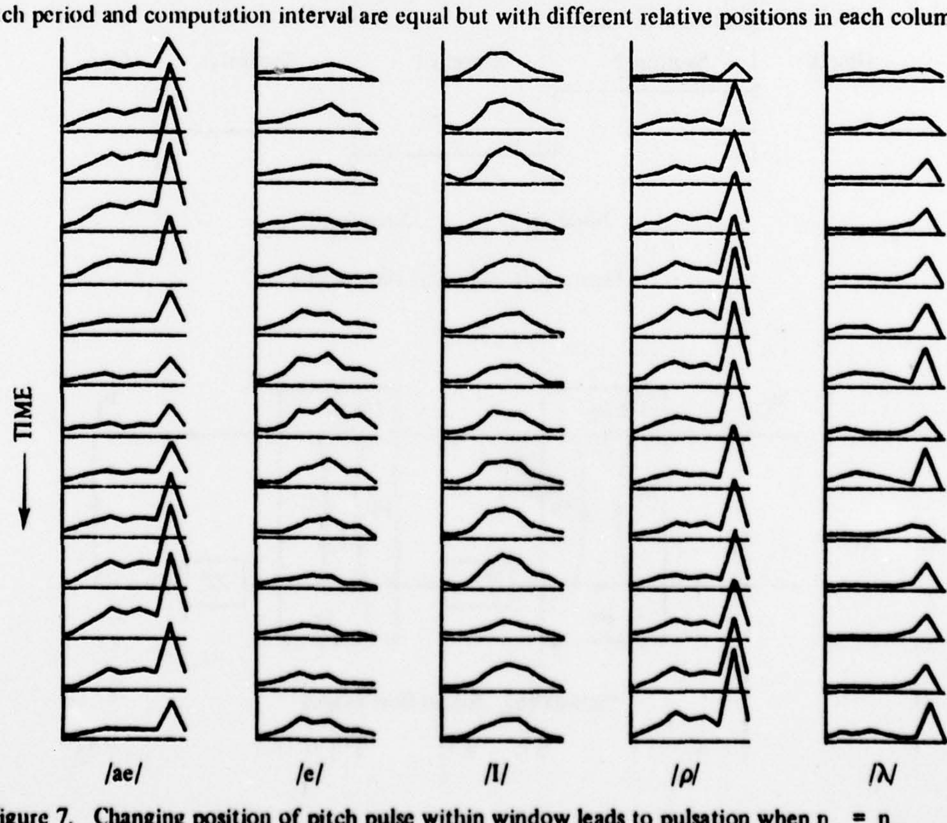

Figure 7. Changing position of pitch pulse within window leads to pulsation when $n_c = n_w$

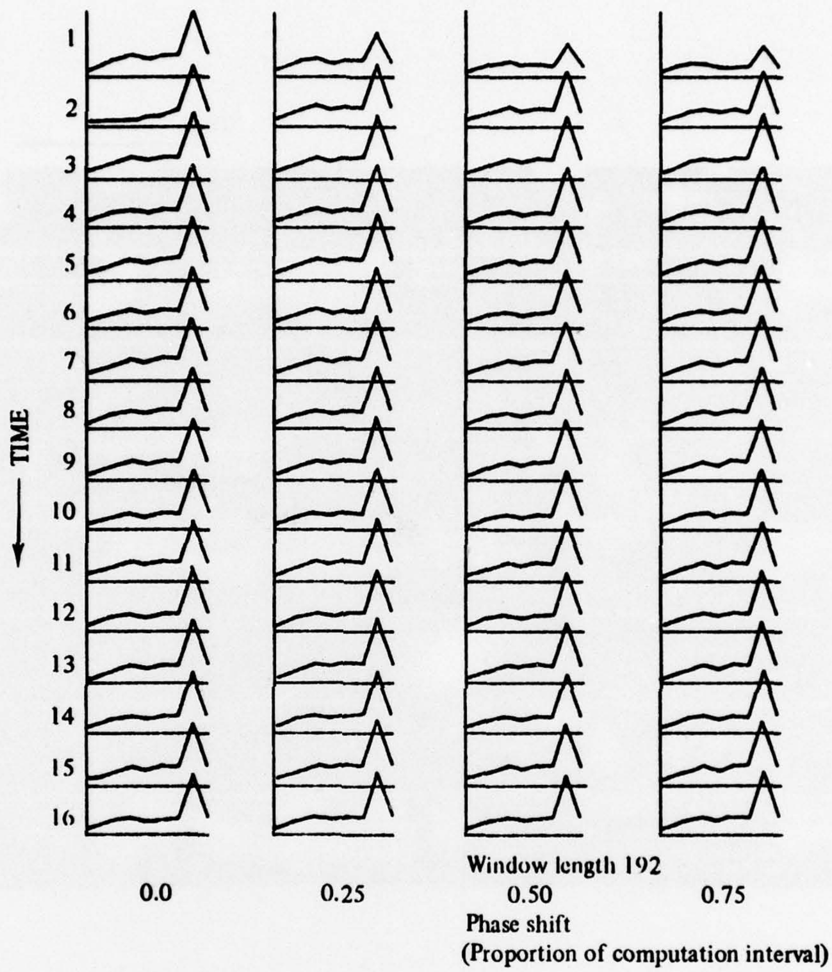

Figure 8. Increasing n_w to 3.5 n_p suppresses the fluctuations, for synthetic vowel /ae/

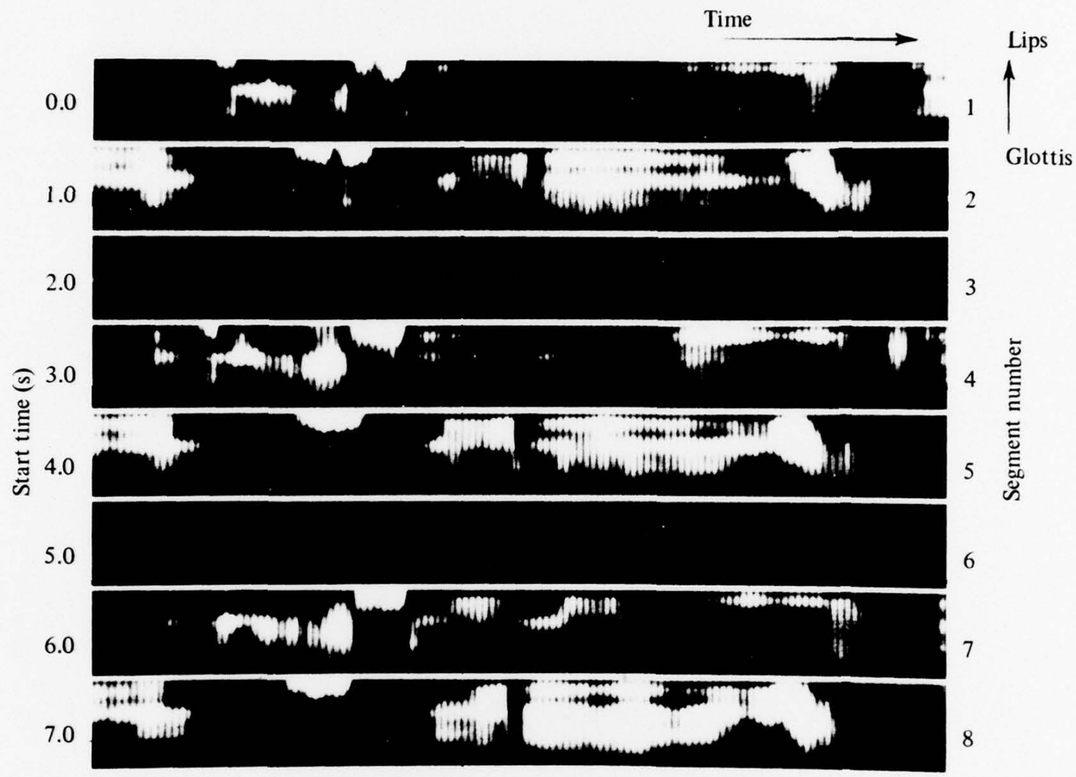

Figure 9. Vocal tract area function for "Speak to me now, bad kangaroo!" for window n_w = 3.5 times pitch period n_p

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